The RCR allows the universal gateway to free the resources (for example, MultiDSP slot card) associated with the VoIP call route that was setup for the call progress tone generation when the first STN was received.

Note If VoIP call persistence was disabled, the RCR would not be needed.

VoIP Call Configuration

3

The voip profile

The voip profile configures call-performance features and manages H.323 call processing. The following VoIP call-performance features are configured through the voip profile:

- The type of voice compression and coding to use for VoIP calls.
- Enabling use of a fixed or dynamic jitter buffer.
- Enabling silence detection and comfort noise generation.
- Adjust the relative lever of silence suppression.
- Modifying the Type-of-Service (ToS) byte for UDP packet processing.
- Modify the maximum number of calls a TAOS unit processes.

The following H.323-specific functions are configured through the voip profile:

- The IP address of the H.323 gatekeeper —MultiVoice Access Manager (MVAM).
- The IP address for a secondary H.323 gatekeeper (MVAM).
- Adjust the frequency and time intervals when a TAOS unit must register with MVAM.
- Enable Personal Identification Number (PIN) collection for authentication by MVAM.
- Enable single-stage dialing.
- Enable progress tone cut-through from the distant PSTN on the local TAOS unit.
- Adjust the amount of time a caller has to dial a telephone number.
- Support for multiple logical gateways.
- Configure voice announcements in place of call-progress tones.
- Enable rerouting of blocked calls back out over the PSTN.

- Enable out-of-band pass through of PSTN call-progress tones.
- Block Caller ID.
- Configure the call interdigit timer.
- Delay alerting the PSTN about active calls until the connection is completed across the MultiVoice network.
- Enable transparent modem signaling.

The MultiVoice call configuration options are located in the `voip` profile:

Parameter	Setting
`voip-index*`	Identifies the `voip` profile by telephone number. This subprofile uses the telephone number (DNIS) associated with an inbound trunk and the called destination to control routing and processing of VoIP calls. The default `voip` profile, `voip { 0 0 }`, is a system wide profile used for processing all VoIP calls. Each DNIS-specific VoIP profile can contain settings which apply only to calls received on the associated trunk.
`gk-mlg-control`	Provides support for partitioning a single MultiVoice Gateway into multiple logical gateways. Partitioning call control lets the H.323 Gatekeeper perform call-specific administration on a call-by-call basis.
`gatekeeper-ip`	Identifies the H.323 gatekeeper associated with this TAOS unit. This gatekeeper, usually running MVAM, performs H.323 registration, admission and status reporting for this MultiVoice Gateway. **Note** After changing the default value of `0.0.0.0` to an IP address, you need to reset the TAOS unit.
`vpn-mode`	Enables or disables H.323 call authentication on a TAOS unit. When authentication is enabled, a TAOS unit prompts for a user- entered Personal Identification Number (PIN).
`packet-audio-mode`	Selects the default audio coder/decoder (codec) used to process analog voice, received from the public switched telephone network (PSTN) and packetized voice, for transmission across the packet network.
`frames-per-packet`	Sets the number of voice frames transmitted in a single RTP packet, across the IP network, between two MultiVoice Gateways.
`tos-options` subprofile	Sets the requested Type-of-Service (ToS) processing priority for RTP packets sent across the IP network between two MultiVoice Gateways.

S 706

Parameter	Setting
silence-det-cng	Setting this parameter to yes enables silence suppression. When enabled, the sending side of the call uses silence suppression during background noise conditions. On the receiving side, suppressed sections will always be filled in with locally generated noise. During those silent periods, the local TAOS unit will generate background (comfort) noise to assure the caller that the call is still connected.
gatekeeper-ip-sec	Identifies a secondary H.323 gatekeeper associated with this TAOS unit. This gatekeeper, usually running MVAM, performs H.323 registration, admission and status reporting for this TAOS unit, if the TAOS unit can't register with the gatekeeper specified by the gatekeeper-ip parameter.
gatekeeper-keepalive	Sets the time interval between attempts a TAOS unit makes to reregister with a system running MVAM, following the initial registration. This value equals the wait time, in seconds, between each attempt to reregister.
registration-retries	Sets the number of attempts a TAOS unit makes each time it executes keepalive registration. Since a MultiVoice Gateway may not successfully register on its first attempt, the value for this parameter represents the number of repeated registration attempts a gateway makes during a registration cycle.
registration-retry-timer	Sets the time interval between each registration attempt a TAOS unit makes with MVAM. This sets the pause, in seconds, between each registration attempt specified by the registration-retries parameter.
primary-retries	Sets the number of attempts a TAOS unit makes whenever it tries to reregister with the MultiVoice Access Manager at gatekeeper-ip after registering with the gatekeeper at gatekeeper-ip-sec. Since a MultiVoice Gateway may not successfully register on its first attempt, the value for this parameter represents the number of repeated registration attempts a gateway makes during a reregistration cycle.
ena-adap-jitter-buffer	Changes the jitter buffer mode to either adaptive or fixed for VoIP calls. When the adaptive mode is selected, the jitter buffer will range in size between the values set for max-jitter-buffer-size and one packet.
max-jitter-buffer-size	Sets the maximum jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using adaptive mode, the jitter buffer may increase to accommodate the entered number of audio packets, based on the incoming packet arrival statistics (jitter).

Parameter	Setting
initial-jitter-buffer-size	Sets the initial jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using either adaptive or fixed mode, the jitter buffer is set to initial-jitter-buffer-size at start-up.
maxcalls	Sets the maximum number of VoIP calls a TAOS unit can process simultaneously, by limiting the number of Digital Signal Processors (DSPs) available.
cut-thru-enable-nearend	Enables or disables transmission of call-progress tones from the far-end public switched telephone network (PSTN) across the IP network to the local TAOS unit, for play out to the caller.
single-dial-enable	Enables or disables single-stage dialing for VoIP calls when MultiVoice is used to perform H.323 call processing.
h323-voice-ann-enabled	Enables or disables play out of voice announcements to report call-progress for VoIP call processing.
voice-ann-dir	Identifies the directory on the external flash memory where voice announcement files are stored for call-progress reporting.
call-inter-digit-timeout	Sets the limit on how long the TAOS unit waits for a caller to enter a single digit when using two-stage dialing, and when entering digits during a call.
silence-threshold	Sets the relative threshold for silence suppression to compensate for background noise levels when silence suppression is enabled (silence-det-cng=yes).
dtmf-tone-passing	Specifies how DTMF tones detected at the ingress gateway are transmitted to the egress gateway. Specify one of the following values: • inband– The near-end gateway passes PSTN-generated DTMF digits and tones as part of the voice processing stream. These tones are compressed by the selected audio codec and transported across the IP network using UDP packets. • outofband–The near-end gateway passes PSTN-generated DTMF digits and tones across the network using non-UDP packets. Once received at the far end, the digits are played out to the local PSTN/caller. • rtp–DTMF tones are transferred and passed via another channel to the decoding DSP, according to the RFC2833 standard. **Note** Both near-end and far-end gateways must have the same setting.

Parameter	Setting
rt-fax-options subprofile	Enables or disables real-time fax call processing.
call-hairpin	Enables or disables attempts to re-route blocked VoIP calls from a TAOS unit using a connection to the local public switched telephone network (PSTN).
call-keep-alive-timeout	Sets the time interval that a MultiVoice Gateway will wait before polling a remote gateway and/or client during a VoIP call, to verify that they are still functioning, and reachable over the IP network.
clid-suppress	Enables or disables blocking transmission of the Calling Line ID (CLID) associated with a call to the local PSTN by the MultiVoice Gateway. The gateway can send a blocked number message or substitute CLID received from MVAM to the PSTN.
true-connect-enable	Enables or disables a TAOS unit delay reporting a call connected to the local PSTN until both parties in a call are connected.
g711-transparent-data	Enables or disables detection of high-speed fax/modem signals on a VoIP channel, and enables fax/modem transmission in a transparent mode using the G.711 codec at 64Kbps.
allow-g711-fallback	Enables or disables fallback to the G.711 audio codec when either H.323 end point (such as, a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the packet-audio-mode parameter.
allow-coder-fallback	Enables or disables fallback to a negotiated codec when either H.323 end point (such as, a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the packet-audio-mode parameter.
trunk-quiesce-enable	Enables or disables automatic trunk deactivation when the MultiVoice Gateway is unavailable to process calls.
early-ringback-enable	Enables or disables local generation of a ringback tone by a TAOS unit as soon as the call-setup begins on the far-end MultiVoice Gateway.
trunk-prefix-enable	Enables or disables assignment of the egress trunk group by the ingress TAOS unit. The MultiVoice Gateway prepends the trunk group number associated with the entry (ingress) T1/E1 trunk to the destination telephone number sent to the exit (egress) gateway or call signaling entity. The egress gateway connects the call to the PSTN using a DS0 assigned to the designated trunk group.

Parameter	Setting
`operator-assist`	Allows callers to request operator assistance during the dialing phase of a MultiVoice call. A TAOS unit can be assigned a dial string, up to five digits long, that may be entered by a caller in order to connect that caller to an operator.
`sequential-call-enable`	If a caller must enter a PIN to authenticate MultiVoice calls, to dial subsequent VoIP calls without reentering the PIN, as long as the connection between the PSTN and near-end MultiVoice Gateway is not terminated.
`next-call`	A new call can be initiated by dialing a string (for example, **9) as specified in the `next-call` parameter in the `voip` profile. Once the dialing string has been entered, the user hears a dial tone and can then proceed to enter the entire 7- or 10-digits (if the call is a long-distance call) number.
`ss7voip-call-persistence`	Setting this to yes causes VoIP call route to persist across VoIP-related IPDC requests for a given call (e.g., LTN, STN, RCCP and RMCP) until the call is released (via RCR).
	Setting this to no disables the feature and the VoIP call route exists only for the life of the single IPDC request, or in the case where an announcement (STN) and DTMF detection (LTN) are overlapping, after the announcement or the DTMF detection has completed, whichever occurs last.
	Enabling VoIP call persistence results in faster call setup and call processing times for SS7 VoIP calls initiated through IPDC.
`faststart-enable`	Setting this to yes results in much faster call setup in the network than that provided by the standard H.245 procedure. In situations in which fast connect is unsuccessful, the call is automatically set up using standard H.245 procedures instead.
`rtpqos-polling-enable`	Setting this to yes generates RTP QoS statistics periodically, through a polling parameter. RTP QoS periodic statistics (such as end-of-call statistics) are sent to the IPDC protocol (this function is not dependent upon the enabling of either RTP QoS polling or Call Logging).
`signaling-tos` subprofile	Enables configuring DSCP values for marking H.323 signaling packets.

S 710

Parameter	Setting
pstn-attribute subprofile	Changes the way an egress MultiVoice Gateway manages call signaling with the switched network. For example: • Delivery of Q.931/Q.850 cause codes are transparent when received from the PSTN by the far-end MultiVoice Gateway to the near-end MultiVoice Gateway. • Bearer capabilities sent in the Q.931 Setup message by the far-end MultiVoice Gateway for outbound calls to the switched network are configurable. • Reporting of Q.931 Progress Indicator information element (IE) in the Proceeding and Alerting message by the near-end MultiVoice Gateway to the switched network is configurable.

If you are configuring a TAOS unit to work in an H.323 environment, you must provide an IP address for the gatekeeper-ip parameter to process VoIP calls. The IP address points to the computer running MVAM that performs all of the H.323 gatekeeper functions for the TAOS unit. The TAOS unit can process VoIP calls over most IP networks using the factory defaults for the remaining voip profile parameters.

Note In this release, you may not change the values contained in DNIS-specific voip profiles. The TAOS unit globally applies the values set in the default voip profile to all VoIP calls. DNIS-specific voip profiles are only used to simplify internal processing and routing of VoIP calls.

Creating DNIS-specific voip profiles

User-defined voip profiles are used to map incoming calls by identifying all calls associated with a specific Dialed Number Identification Service (DNIS) string as VoIP calls. See "Using DNIS-specific trunk mappings" on page 2-15.

For example, if a user created the following voip profiles:

```
admin> dir voip
    46  12/23/1998 09:48:55  { 0 0 }
    31  12/18/1998 09:50:06  { 8093190 0 }
    31  12/18/1998 10:07:16  { 8903190 0 }
```

The TAOS unit processes all calls from the PSTN with these DNIS strings as VoIP calls. The voip-index subprofile distinguishes between the default voip profile, voip {0 0}, and any user-created voip profiles:

```
admin> list voip-index
[in VOIP/{ 8903190 0 }:voip-index
gateway-access-number = 8903190
far-end-number = 0
```

This subprofile includes the following parameters:.

Parameter	Specifies
gateway-access-number	This is the Dialed Number Identification String (DNIS) passed from the PSTN associated with the in-bound telephone number used to access the TAOS unit. If the TAOS unit is configured to perform two-stage dialing of VoIP calls, this would be the telephone number dialed to access the TAOS unit from the PSTN.
far-end-number	This value should always be set to 0.

To set the values for these parameters, use the new and write commands to create user-defined voip profiles:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> new voip { 8903190 0 }
VOIP/{ 8903190 0 } read
admin> write
VOIP/{ 8903190 0 } written
admin> dir voip
   46 12/23/1998 09:48:55 { 0 0 }
   31 12/18/1998 10:07:16 { 8903190 0 }
```

Note You may create DNIS-specific voip profiles by changing the value of the gateway-access-number parameter using the set command; but only if no other changes have been written to the voip { 0 0 } profile.

Configuring call-performance parameters

The call-performance parameters control how a MultiVoice Gateway processes calls received from the PSTN. This group of parameters affects allocation of packet network bandwidth for each call, the allocation of DSP assets for each call, and subsequently, the number of VoIP calls that a TAOS unit can process simultaneously. The following voip profile parameters handle VoIP call-performance functions:

- packet-audio-mode
- frames-per-packet
- allow-g711-fallback
- allow-coder-fallback
- silence-det-cng
- silence-threshold
- ena-adap-jitter-buffer
- max-jitter-buffer-size
- initial-jitter-buffer-size
- tos-options
- maxcalls
- faststart-enable

Configuring voice compression

Voice is transmitted across an IP network as compressed audio frames, which are compressed/decompressed by the TAOS unit. The `packet-audio-mode` parameter specifies which default audio codec (coder/decoder) packs (and unpacks) analog speech into digital audio frames. You may enter any of the following values representing these supported audio codecs:

Parameter value	Specifies
g711-ulaw	G.711 µ–law
g711-alaw	G.711 a-law
g729	G.729(A)
g723	G.723.1 running at 5.3kps
g723-6.4kps	G.723.1 running at 6.4kps
g728	G.728
frgsm	Full-rate GSM

The default value for the `packet-audio-mode` parameter is `g711-ulaw`. The following example illustrates how to set the audio codec used for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set packet-audio-mode = g729
admin> write
VOIP/{ 0 0 } written
```

Changes to the setting take effect with the next call. The `packet-audio-mode` parameter has the following dependencies:

- TAOS units configured with 96-port MultiDSP slot cards (APX8-SL-96DSP) support using only the G.711 or G.729(A) audio codecs.
- This parameter does not prevent other supported audio codecs from being dynamically selected during call-setup.
- The `silence-det-cng` parameter is ignored when using G.711 a-law or G.711 µ–law. For details, see "Configuring silence detection and comfort noise generation" on page 3-13.
- When either G.723 or G.723-6.4kps codec is specified:
  - `silence-det-cng` may be enabled or disabled for 6.4Kbps processing only (`packet-audio-mode=g723-6.4kps`).
  - Comfort noise generation may be enabled or disabled for 5.3Kbps processing. With comfort noise enabled, the 5.3Kbps can decode silence detection and suppression packets. Silence detection/suppression cannot be selected for 5.3Kbps processing since it will not encode silence. This is in accordance with standards.
  - Comfort noise generation cannot be enabled for 5.3Kbps processing unless the adaptive jitter buffer is disabled.
  - Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

- Adaptive jitter buffer processing can be enabled when silence detection/suppression is disabled.
- The actual maximum size of the adaptive jitter buffer is limited to nine frames per packet for G.723.1 both rates.

G.728 codec support

G.728 is an audio codec based on Low-Delay Code Excited Linear Prediction (LD-CELP). G.728 provides toll-quality audio at a bit rate of 16Kbps. With a frame size of only 2.5 milliseconds, G.728 also has a very low delay. Although the MultiVoice implementation of G.728 uses a frame size of 5 milliseconds, the bitstream from the audio codec is the same as described in the ITU-T standard and can thus be decoded by any G.728 decoder.

When the G.728 codec is selected, the MultiVoice Gateway attempts to determine if the G.728 codec is supported by the other gateway during H.245 capability negotiation. If both sides agree to use G.728 as the preferred codec, both gateways use G.728 to compress and decompress audio after the H.245 open logical channel message is exchanged.

Although MultiVoice uses a 5-millisecond frame for G.728 processing, it is compatible with any third-party G.728 decoder. However, if a MultiVoice Gateway attempts to communicate with a third-party VoIP gateway transmitting an odd number of 2.5 millisecond frames per IP packet, the call fails.

When you enable G.728 audio processing (`packet-audio-mode=g728`), the `silence-det-cng` parameter in the `voip` profile must be set to no (its default value). The following commands enable G.728 processing:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-audio-mode = g728

admin> set silence-det-cng = no

admin> write
VOIP/{ 0 0 } written
```

G.723.1 codec support

When G.723.1 codec is selected

- `silence-det-cng` can be enabled or disabled for 6.4Kbps processing only by setting the `packet-audio-mode` parameters as follows:

  `packet-audio-mode=g723-6.4kps`
- Comfort noise generation can be enabled or disabled for 5.3Kbps processing by setting the `packet-audio-mode` to `g723-5.3kps`. When comfort noise is enabled, the 5.3Kbps setting allows silence detection and suppression packet decoding. Silence detection/suppression cannot be configured directly for 5.3Kbps processing since it will not encode silence. This is in accordance with standards.
- Comfort noise generation cannot be enabled for the `packet-audio-mode` 5.3Kbps setting processing unless adaptive jitter buffering is disabled and the `ena-adap-jitter-buffer` is set to no.
- Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

S 714

- Adaptive jitter buffer processing can be enabled for:
  - 6.4Kbps processing when silence detection/suppression is disabled.
  - 5.3Kbps processing when comfort noise generation is disabled.
- The actual maximum size of the adaptive jitter buffer is limited to nine frames per packet for G.723.1 both rates.

Full-Rate GSM codec support

Full-Rate GSM (Global System for Mobile Communications) is a voice encoder/decoder standard for cellular communications. Full-Rate GSM compresses the speech samples from 64Kbps PCM to 13.2Kbps, requiring less network than G.711 a-law or G.711 µ–law. It is the standard followed by European, Japanese, and Australian cellular communications systems, and is supported by certain Web phone applications.

Full Rate GSM uses a speech frame size of 160 samples (20msec) and the encoder produces 33 bytes per frame. The decoder produces 160 samples (20msec) of speech from the 33-byte encoder output.

The Full Rate GSM audio codec is defined by ETSI Recommendation GSM 06.10, *GSM Full Rate Speech Transcoding*, (Feb. 1992), European Telecommunications Standards Institute. Full Rate GSM also supports Silence Detection and Comfort Noise Generation, as defined by the ETSI Recommendation GSM 06.12, *Comfort Noise Generation*, (Feb. 1992), European Telecommunications Standards Institute and ETSI Recommendation GSM 06.12, *Discontinuous Transmission* (Feb. 1992), European Telecommunications Standards Institute.

A MultiVoice Gateway reports Full-Rate GSM during H.245 capability negotiation. If both H.323 end points (such as, a MultiVoice Gateway and a PC, or two MultiVoice Gateways) choose Full-Rate GSM as the preferred codec, then, after opening the H.245 logical channel between both H.323 end points, Full-Rate GSM is used for processing the VoIP call. Full-Rate GSM is encoded as a standard audio capability.

Configuring voice packet size

The number of compressed audio frames assigned to each RTP packet used to transport voice across the IP network is controlled by the `frames-per-packet` parameter. You can assign a value ranging from 1 to 10 packets. The default is 4.

The following example illustrates how to change the number of audio frames assigned to each RTP packet for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set frames-per-packet = 6
admin> write
VOIP/{ 0 0 } written
```

Assigning a lower value to the `frames-per-packet` parameter reduces the delay and distortion introduced into any given voice call. But using a lower value may also degrade performance as the number of RTP packets processed for a voice call increases.

Note When a different audio codec is dynamically selected during call-setup, a TAOS unit uses the default value of four frames per RTP packet to process that call.

For more information on MultiVoice packet processing see Appendix A, "MultiVoice Packet Processing."

Configuring audio codec negotiation

Voice is transmitted across an IP network as compressed audio frames. The `packet-audio-mode` parameter in the default `voip` profile specifies the preferred audio codec used by the gateways to compress and uncompress analog speech and digital audio frames.

You can use the following parameters (shown with default values) to specify how the system behaves when the preferred codec is not supported for all VoIP, fax, and transparent modem calls:

```
[in VOIP/{ 0 0 }]
allow-g711-fallback = yes
allow-coder-fallback = yes
```

Parameter	Setting
`allow-coder-fallback`	Overrides fallback to a negotiated codec when either of H.323 end points (such as a gateway or terminal) involved in a VoIP call do not support the audio codec designated by the `packet-audio-mode` parameter. When `allow-coder-fallback`=no, the ingress gateway rejects the call if it is unable to connect the call using the preferred codec. If this parameter is set to no, the `allow-g711-fallback` parameter has no effect. When `allow-coder-fallback`=yes, the ingress gateway negotiates an alternate audio codec with the destination gateway during call capabilities setup. Yes is the default setting.
`allow-g711-fallback`	Overrides fallback to the G.711 audio codec when either of H.323 end points (such as a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the `packet-audio-mode` parameter. If `allow-coder-fallback`=yes, setting `allow-G711-fallback`=no prevents the ingress gateway from selecting the G.711 codec when negotiating call capabilities. In this case, the system terminates the call if G.711 is the only available choice and it is not the preferred codec. When `allow-G711-fallback`=yes, the ingress gateway may negotiate using the G.711 audio codec with the destination gateway during call capabilities setup. yes is the default setting.

Normally, an H.323 stack advertises a list of supported audio codecs. If the preferred codec is present on a list received from a far-end gateway, that codec is always selected. Otherwise, the system selects an alternate codec that matches one from its supported list.

Modifications made to these parameters become effective with the next VoIP call.

The following example illustrates how to allow fallback to any supported audio codec, except G.711, when processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set allow-coder-fallback = yes
admin> set allow-g711-fallback = no
admin> write
VOIP/{ 0 0 } written
```

Configuring silence detection and comfort noise generation

A TAOS unit can be configured to detect periods of silence during voice calls, suppress transmission of voice packets containing silence, and generate white (comfort) noise to assure the user that a call is still connected during silent periods.

The silence-det-cng parameter enables or disables the feature. You can prevent silence frames from being passed across the packet network, reducing the effective bandwidth of the VoIP call. During those silent periods, the local TAOS unit generates background (comfort) noise to assure the caller that the call is still connected during these silent periods. You can assign the following values to the silence-det-cng parameter:

Parameter value	Specifies
yes	Silence frames are not passed across the IP network by the TAOS unit. During silent periods, while the call is still connected, the local TAOS unit will generate background (comfort) noise.
no	(Default) Silence is processed as part of the audio stream; comfort noise is not locally generated.

Changes to the parameter setting become effective with the next VoIP call.

The following example illustrates how to enable silence detection and comfort noise generation when processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set silence-det-cng = yes
admin> write
VOIP/{ 0 0 } written
```

When using silence suppression and comfort noise generation, the following apply:

- Silence compression and comfort noise generation must be enabled on both the local TAOS unit and distant TAOS unit involved in a call.
- When silence compression and comfort noise generation are enabled, the dynamic jitter buffer is not used (ena-adap-jitter-buffer=no).
- When either the G.723 or G.723-6.4kps codec is specified

S 717

- Comfort noise generation can be enabled for processing at a rate of 5.3Kbps. With comfort noise enabled, the 5.3Kbps processing can decode silence detection and suppression packets.
- Comfort noise generation cannot be enabled for 5.3Kbps processing unless the adaptive jitter buffer is disabled.
- Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

Adjusting the relative silence threshold

The `silence-threshold` parameter adjusts the relative threshold for silence suppression to compensate for background noise levels. This parameter lets the silence floor be raised or lowered in 1dB increments, without preventing conversation at normal speech levels from getting through. This parameter allows the user to raise the silence floor from an increase of 0 dB, the nominal level, to an increase of 9dB.

The following example illustrates how to raise the relative threshold for silence suppression for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set silence-threshold = 3
admin> write
VOIP/{ 0 0 } written
```

Dependencies This parameter is ignored if silence suppression is disabled (`silence-det-cng=no`).

Configuring dynamic call jitter buffers

VoIP calls are processed using packet-based jitter buffering. A unique jitter buffer is opened for each call; the buffer dynamically adjusts its size to accommodate network jitter. In essence, jitter buffer playout delay adapts to network jitter.

To configure dynamic call jitter buffers, proceed as follows:

1. **Enable the adaptive jitter buffer.**
2. **Configure the initial jitter buffer size.**
3. **Configure the maximum jitter buffer size.**

Enabling the adaptive jitter buffer

The `ena-adap-jitter-buffer` parameter changes the jitter buffer mode to either adaptive or fixed for VoIP calls. When the adaptive mode is selected, the jitter buffer ranges in size between the values set for `max-jitter-buffer-size` and one packet, depending on the number of late or out-of-sequence packets received during the call. You can enter either of the following values:

Parameter value	Specifies
yes	(Default) That adaptive jitter buffering is used for processing VoIP calls.

Parameter value	Specifies
no	That static jitter buffers is used for processing VoIP calls.

Changes to this value become effective with the next VoIP call.

The following example illustrates how to enable or disable adaptive jitter buffering for VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set ena-adap-jitter-buffer = yes
admin> write
VOIP/{ 0 0 } written
```

Using adaptive jitter buffering has the following dependencies:

- When silence-det-cng=yes, MultiVoice uses the value assigned to initial-jitter-buffer-size parameter to open static call jitter buffers.
- When ena-adap-jitter-buffer=no, MultiVoice uses the value assigned to initial-jitter-buffer-size to open static call jitter buffers.
- When G.723 codec is the valued for the packet-audio-mode parameter, the maximum jitter buffer size can't exceed nine packets (max-jitter-buffer-size=9).

Configuring the maximum jitter buffer size

The max-jitter-buffer-size parameter sets the maximum jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using adaptive mode, the jitter buffer can increase to accommodate the entered number of audio packets, based on the incoming packet arrival statistics (jitter).

You may enter a value between 1 and 19 (packets). This allows the TAOS unit to expand the length of a call's jitter buffer to a size proportionate to the selected number of audio packets. This value defaults to 19. Changes to this value become effective with the next VoIP call.

The following example illustrates how to set the maximum jitter buffer size when adaptive jitter buffering is enabled for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set max-jitter-buffer-size = 19
admin> write
VOIP/{ 0 0 } written
```

Configuring the initial jitter buffer size

The initial-jitter-buffer-size parameter sets the initial jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using either adaptive or fixed mode, the jitter buffer is set to initial-jitter-buffer-size at start-up. During a call, the TAOS unit adjusts each jitter buffer to accommodate the number of audio packets, based on the in-coming audio packet volume.

You can enter a value between 1 and 19 (packets). This value defaults to 2. Changes to this value become effective with the next VoIP call.

Note When using adaptive jitter buffers, the minimum jitter buffer size may be less then the value assigned to the `initial-jitter-buffer-size` parameter. Under the appropriate conditions, adaptive jitter buffers may shrink to only one frame in size from the `initial-jitter-buffer-size`.

The following example illustrates how to change the initial jitter buffer size when adaptive jitter buffering is enabled for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set initial-jitter-buffer-size = 5
admin> write
VOIP/{ 0 0 } written
```

For more information on jitter buffer processing see Appendix B, "Determining Jitter Buffer Size."

Type of Service (TOS) or Differentiated Service Codepoint (DSCP) marking

You can set the IP Type of Service (TOS) byte in IP packets that carry signaling messages to define the type of packet marking— either TOS or Differentiated Services Codepoint (DSCP).

For detailed information about how the system supports TOS precedence (RFC 791) and DSCP (RFC 2474) marking of packets, see the *APX/MAX TNT WAN, Routing and Tunneling Configuration Guide.*

Type of Service marking

Type of Service is an eight-bit parameter found in the header of an IP datagram. In networks that support processing of IP packets based on precedence, the Type of Service byte is used to attain a certain level of UDP packet processing by manipulating values for delay, throughput, and reliability.

The `tos-options` subprofile sets the Precedence bits (bit0 - bit2) and the TOS bits (bit3 - bit6) for the Type of Service (TOS) byte use by UDP voice packets. It is divided into three fields, containing the following values:

Bits 0-2: Precedence.

Bits 3-6: TOS (performance cost).

Bit 7: Reserved for Future Use.

0	1	2	3	4	5	6	7
PRECEDENCE			TOS				0

The `tos-options` subprofile is shown below, with default values:

```
[in VOIP/{ 0 0 }:tos-options]
active = yes
precedence = 101
type-of-service = latency
apply-to = both
```

Parameter	Setting
active	Enables or disables user configuration of the Type-of-Service byte. Setting this value to yes enables operator configuration of the TOS byte. This is the default value. Setting this value to no disables this feature. Changes to this parameter take effect with the next VoIP call.
precedence	Importance of priority of the UDP packet, bit0 through bit2 of the Type-of-Service octet. This is represented by a hexadecimal value, which defines how the network processes the UDP packets. The default is 101. Changes to this value become effective with the next call.
type-of-service	Processing attribute management, bit3 through bit6 of the Type-of-Service octet. These bits denote how the network should make trade-offs between throughput, delay, reliability and cost when processing the UDP packets. This value defaults to latency. Changes to this value become effective with the next call.
apply-to	How the Type-of-Service value is applied to the data flow over the IP network between the MultiVoice Gateways. This parameter has no affect on VoIP call packet processing.

Configuring precedence parameter values

The values assigned to the precedence parameter set bit0 through bit2 of the Type-of-Service octet. The impact of a selected value on UDP packet processing is IP network dependent (see RFC791). You can enter of the following values (hexadecimals), representing processing priorities, as defined by RFC791:

Parameter value	Specifies (RFC 791 definition)
000	Routine
001	Priority
010	Immediate
011	Flash
100	Flash Override
101	CRITIC/ECP (default)
110	Internetwork Control
111	Network Control

The following example illustrates how to assign a “Flash” precedence or processing priority to UDP packets used for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> list tos-options
```

```
admin> set precedence = 011
admin> write
VOIP/{ 0 0 } written
```

Configuring type-of-service parameter values

The values assigned to the type-of-service parameter set bit3 through bit6 of the Type of Service octet. The impact of a selected value on UDP packet processing is network dependent (for more information, see "Use of the TOS field in Routing" in RFC1349):

Parameter value	Specifies (RFC1349 definition)
latency	Minimize delay
throughput	Maximize throughput
reliability	Maximize reliability
cost	Minimize cost
normal	Normal (network control)

The following example illustrates how to set maximized throughput for processing UDP packets used for VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> list tos-options
admin> set type-of-service = throughput
admin> write
VOIP/{ 0 0 } written
```

Differentiated Services Codepoint marking

You can use the IP TOS field in the IP header of packets that carry H.323 signaling messages to set DSCP.

Differentiated services (DS) is an architecture that provides different types or levels of service for network traffic. The differentiated services codepoint is a particular bit pattern (that is, a hexadecimal value) that can be assigned to the DSCP 6-bit field in the IP TOS byte of the IP header. This DSCP field facilitates the definition of future per-hop behaviors.

Implementors should note that the DSCP field is six bits wide. Differentiated Services compliant nodes must select per-hop behaviors by matching against the entire 6-bit DSCP field.

Note The full byte (that is, 8 bits) of the DSCP field can be specified and will be set this way in the IP TOS byte of the IP header. Even though you can set the entire 8 bit in any desired way, only the most significant 6 bits are used and matched to select a Per-Hop Behavior (PHB) by the DS domains in the network. In order to specify traditional TOS/Precedence values, as per RFC 791, the desired bit field can simply be specified as the equivalent DSCP value.

Parameters in voip profile sets DSCP in H.323 packets

You can configure DSCP values for marking H.323 signaling packets by setting the following parameters in the signaling-tos subprofile of the voip profile:

```
[in VOIP/{ 0 0 }:signaling-tos]
active = no
precedence = 000
type-of-service = normal
apply-to = both
marking-type = dscp
dscp = 00
```

Parameter	Setting
active	Enables or disables user configuration of the DSCP. Setting this value to yes enables configuration of the DSCP. This is the default value. Setting this value to no disables this feature. Changes to this parameter take effect with the next VoIP call.
precedence	Importance of priority of the UDP packet, bit0 through bit2 of the Type-of-Service octet. This is represented by a hexadecimal value, which defines how the network processes the UDP packets. The default is 000, which means Normal Priority. Changes to this value become effective with the next call.
type-of-service	Processing attribute management, bit3 through bit6 of the Type-of-Service octet. These bits denote how the network should make trade-offs between throughput, delay, reliability and cost when processing the UDP packets. This value defaults to normal. Changes to this value become effective with the next call.
apply-to	How the Type-of-Service value is applied to the data flow over the IP network between the MultiVoice Gateways. This parameter has no affect on VoIP call packet processing.
marking-type	Either precedence-tos or dcsp. When set to dscp, Differentiated Services CodePoint marking (RFC 2474) can be set by entering a hexadecimal number via the dscp parameter. The differentiated services codepoint is a particular bit pattern (that is, a hexadecimal value) that can be assigned to the Differentiated Services Codepoint (DSCP) 6-bit field in the IP TOS byte of the IP header. When set to precedence-tos, the system marks packets in a manner consistent with RFC 791.

Parameter	Setting
dscp	The DSCP tag to be used in the marking of the packets (if the `marking-type` parameter = dscp). Hexadecimal field, 1 byte. The default value is 00 and the range is from 00 to `ff` hexadecimal.

For example, the following commands enable DSCP marking and specify a value of 33 for H.323 signaling packets:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set signaling-tos active = yes
admin> set signaling-tos dscp = 33
admin> write
```

For details about configuring DSCP marking in SS7 signaling packets, refer to your platform's *Physical Interface Configuration Guide*.

Controlling VoIP call volume

The `maxcalls` parameter controls the maximum number of VoIP calls a TAOS unit can process simultaneously, by limiting the number of digital signal processors (DSPs) available for processing VoIP calls.

Note When the `voip-max-capacity-allowed` parameter is enabled and licensed for an APX in the read-only base profile, the `maxcalls` parameter is automatically set to the maximum number of VoIP calls that can be processed (for example, 2688). (See "Base profile parameters" on page 2-2 for details.)

Limited DSP availability is useful when continued high call volumes on a network affect the call quality. Adjusting the value for `maxcalls` allows a TAOS unit to allocate more system resources to processing fewer calls, resulting in improved call quality. When active calls exceed the resources that are available to process VoIP calls, the caller hears a busy signal from the TAOS unit.

Maxcalls parameter

The following example illustrates how to limit the number of available DSPs to handle VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set maxcalls = 128
admin> write
VOIP/{ 0 0 } written
```

You may enter any number between 1 and the maximum value. On a MAX TNT, the `maxcalls` parameter defaults to 672—only values between 1 and 672 can be entered. On a APX, only values between 1 and 2688 can be entered. If an APX has only been licensed for up to 2688 calls, and it receives call attempt #2689, the call will be rejected.

Changes to this parameter become effective with the next call.

Note This value does not reflect actual VoIP call volumes achieved by the TAOS unit in either a testing or production environment.

Exceeding the maximum call volume

When the licensed call volume is exceeded, a MultiVoice Gateway displays the following warning message if debugging and the h323warn command (level 2 or higher) were enabled:

```
H323:    4: WARNING: _wanNewCall():
... call denied due to simultaneous call capacity
... 2689 > 2688
```

Configuring H.323 (v2) fastStart

The H.323 (v2) fast connect procedure allows for faster call completion. Fast connect provides faster call setup and with fewer round-trip connections needed to establish a call between end points.

H.323 (v2) defines a fast connect procedure, which is also known as *fastStart*. This fast connect procedure streamlines the connection establishment of calls when:

- Capabilities exchange is not necessary.
- End point compatibility is assumed.

H.245 capabilities exchange is performed *after* the fast connect procedure is completed, because the logical channel set-up exchange is embedded in the H.225 message exchange. However, open logical channel exchange is not performed.

With fast connect, messaging can be collapsed into a single handshake consisting of a setup message and a connect message.

fastStart vs. standard H.245 procedure

The fast connect procedure results in much faster call setup in the network than that provided by the standard H.245 procedure. In situations in which fast connect is unsuccessful, the call is automatically set up using standard H.245 procedures instead.

Upon completion of the fast connect procedure, to set up a voice call, the H.245 procedure is initiated and all mandatory H.245 procedures need to be completed using either H.245 tunneling or H.245 connection. This is especially important if you use a third-party gateway that does not support the fallback condition. In this case, the call will be released due to H.245 time-out.

H.323 (v2) fast connect call flow

The following call flow occurs when H.323 (v2) fast connect is used.

The calling end point sends a setup message to the called end point. The setup message contains a fastStart element with the following audio mode information:

- Codec
- Rate
- RTP/RTCP addresses

If the called end point initiates the use of the fast connect procedure for the call, the called end point may return information in the call proceeding, call alerting, and call connect messages that contain a fastStart element.

If the called end point fails to initiate the use of the fast connect procedure, the called end point may respond with a call proceeding, call alert or call connect message that does not contain a fastStart element.

If the calling end point receives call proceeding, call alert, or call connect messages without a fastStart element, the calling end point terminates the fast connect procedure. The calling end point also completes the H.245 procedure, using one of the following two methods:

- H.245 tunneling, provided that H.323 tunneling is supported at both end points.
- A separate H.245 channel.

Reverting to the H.245 connection

When fast connect is being used, either end point can initiate a separate H.245 connection at any time. Initiation of an H.245 connection is required under either of the following conditions:

- If either end point does not support the fastStart element and H.245 tunneling.
- If a call uses the fastStart element and if H.245 tunneling is not supported for the call.

When either end point initiates a separate H.245 connection, this supports:

- Fax transmission.
- Invoking the call feature that require the use of H.245 procedures such as Out-of-Band (OOB) DTMF.

H.245 call flow

All mandatory H.245 protocol elements that normally occur upon initiation of an H.245 connection are completed prior to initiation of any additional H.245 procedures. These include:

- Cap exchange
- Master/slave determination

Note The media channels that are established as a result of the fast connect procedure are inherited as though they had been opened using normal H.245 `OpenLogicalChannel` and `OpenLogicalChannelAck` procedures. For such inheritance to succeed, media sessions opened during the fast connect procedure must use only well-known sessionID values, as defined in the H.245 standard.

Using fastStart with H.245 tunneling

When a fastStart element is being used, either end point can initiate the use of H.245 tunneling. H.245 tunneling is required under either of the following circumstances to:

- Support the fax transition.
- Invoke call features that require the use of H.245 procedures.

A calling end point can also include both a fastStart element and can set the `h245Tunneling` field to `TRUE` within the same setup message. Similarly, a called end

point can include a fastStart element and set the `h245Tunneling` field to `TRUE` within the same Q.931 response. In this instance, the fast connect procedures are followed, and the H.245 connection is not established until the actual transmission of the first tunneled H.245 message has occurred, or until the separate H.245 connection has been opened.

Note In the H.323 (v2) standard, the calling end point must include one but *not* both of the following in the same setup message:

- A fastStart element.
- An encapsulated H.245 messages in H245Control.

The presence of the encapsulated H.245 message in this instance overrides the Fast Connect procedure.

Terminating the H.323 V2 Fast Connect Procedure

The Fast Connect procedure is terminated when one of the following events has occurred:

- An encapsulated H.245 message is sent.
- A separate H.245 connection by either end point prior to the sending of a Q.931 message containing fastStart by the called end point is initiated.

faststart-enable parameter

The `faststart-enable` parameter enables and disables the fastStart feature. If the `faststart-enable` parameter is enabled (set to yes), the fast connect procedure is initiated. yes is the default value.

The following procedure illustrates how to enable fastStart:

```
[in VOIP/{ 0 0 } read]
admin> set faststart-enable=yes
admin> wri
```

Configuring H.323 call management parameters

A TAOS unit may be configured to use IPDC in support of an SS7 network configuration or use H.323 in support of non-SS7 networks. TAOS units use the following parameters to handle H.323 management:

- `gatekeeper-ip`
- `gatekeeper-ip-sec`
- `gatekeeper-keepalive`
- `registration-retries`
- `registration-retry-timer`
- `primary-retries`
- `cut-thru-enable-nearend`
- `dtmf-tone-passing`
- `clid-suppress`
- `call-keep-alive-timeout`
- `call-inter-digit-timeout`

- `true-connect-enable`
- `single-dial-enable`
- `call-hairpin`
- `vpn-mode`
- `h323-voice-ann-enabled`
- `voice-ann-dir`
- `voice-ann-enc`
- `g711-transparent-data`
- `trunk-quiesce-enable`
- `early-ringback-enable`
- `trunk-prefix-enable`
- `rtpqos-polling-enable`

These parameters can be ignored or reset when using IPDC to route VoIP calls from an SS7 network. When operating in an H.323 environment, the only parameter that must be set is the `gatekeeper-ip` parameter. This identifies the location of the H.323 gatekeeper system that performs call management for the MultiVoice network.

Note The use of the H.323 `voice-ann-enabled`, `voice-ann-dir`, and `voice-ann-enc` parameters is discussed in Chapter 4, "Voice Announcement Administration." The use of the `rt-fax-options` subprofile is discussed in Chapter 5, "MultiVoice Real-time Fax."

H.323 gatekeeper communication

An H.323 gatekeeper manages the MultiVoice network. It provides address translation and controls access to the local area network for all TAOS units processing VoIP calls and any H.323 terminals (such as, PCs running H.323-compliant telephony software). All gatekeeper functions are performed by the MultiVoice Gatekeeper running MVAM software.

Identifying the primary gatekeeper

The `gatekeeper-ip` parameter identifies the computer running MVAM that performs all the H.323 gatekeeper functions for this TAOS unit when MultiVoice is configured to perform H.323 call processing.

MultiVoice implements the H.323 direct call model for VoIP networks, so each TAOS unit must communicate with a gatekeeper to perform call registration and admission, and report statuses (RAS). The TAOS unit sends all call request messages and call processing information to the IP address specified by `gatekeeper-ip`.

After changing the default value of `0.0.0.0` to an IP address, you need to reset the TAOS unit. Changes to this parameter take effect with the next registration cycle.

Note The `gatekeeper-ip` parameter must be configured for a TAOS unit to start processing VoIP calls in an H.323 network.

The following example illustrates how to set the IP address of MVAM that functions as the H.323 gatekeeper for this TAOS unit:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
```